INFORMATION ABOUT USE OF THE SIGNALOGIC REAL-TIME CONVOLVER

OVERVIEW

The following instructions assume that you have a working familiarity with a Hypersignal-Macro family package (-Macro, -Macro EX, or -Acoustic), and know how to use basic Hypersignal menus, including the Macro Manager

menu. The current Real-Time Convolver version is 1.6, January 1998. The copyright in all DSP and macro source code belongs to Signalogic; all rights are reserved. In the instructions below, the terms "filter" and "impulse response" are used interchangeably.

The Real-Time Convolver consists of executable DSP files and a Hypersignal macro. Currently, the following DSP files are included:

rtc32C.out	DSP file for DSP32C-based systems
rtcx32C.out	DSP file for DSP32C-based systems that use memory mode 5
rtc3210.out	DSP file for DSP3210-based systems
rtcC30.out	DSP file for TMS320C3x-based systems (not yet available)
rtcC40.out	DSP file for TMS320C4x-based systems (not yet available)

Currently, the following macro files are included:

rtc.mac	main real-time convolution program file
f_rtc.mac	real-time convolution function file
rtc.mcf	macro configuration file

You must have Hypersignal-Acoustic, Hypersignal-Macro EX, or Hypersignal-Macro real-time software, version 3.40f or higher, in order to run the Real-Time Convolver. Although the Real-Time Convolver depends upon a Hypersignal-family package to download an executable DSP file and the impulse response coefficients to the target DSP board, you should not assume that the convolver supports all hardware supported by Hypersignal, because the DSP code used by the convolver is completely separate and unrelated to that used by Hypersignal packages. Also, if you are using hardware only supported by Hypersignal-Acoustic, such as an Ariel PC-32C with ProPort or DatPort, then this limitation (operation with Hypersignal-Acoustic only) remains in effect.

TO INSTALL THE REAL-TIME CONVOLVER

Copy all files from the Real-Time Convolver distribution diskette to the subdirectory where your Hypersignal .exe file and other Hypersignal files are located.

TO RUN THE REAL-TIME CONVOLVER

Below are instructions to run the Real-Time Convolver:

1. First, it is assumed that you have your current DSP/Analog hardware installed, configured in the ANALOG CONVERSION field of the Hypersignal System Configuration menu, and working. For example, if you are

using an Ariel DSP-32C board, then you should have 'ARI32C' entered in the ANALOG CONVERSION field, and you should have tested the overall system at least once by acquiring a test file of analog input data to disk. Such a test is imperative, because the Real-Time Convolver does not check to determine whether DSP/Analog boards are properly installed, responding to current I/O or memory address jumpers, and adjusted for current signal input levels.

2. Second, invoke the Macro Manager menu (press the F3 key from the Hypersignal main manu), and enter 'rtc' in the first field (filename field). Enter the filename ***only***; no extension is necessary.

3. After entering the macro name and pressing the 'Enter' key, you should see the following macro input fields:

   Channel List Enter 0 for channel 0 only, or 0,1 for channels 0 and 1.

   Filter Name 1 Enter the name of the filter or impulse response file you with which you wish to convolve real-time input data. The filename can represent any data in Hypersignal FIR coefficient file format or Hypersignal time domain waveform file format. In both cases, a basic FIR filter model is assumed, with successive coefficients or samples assumed to be taps of the filter.

   Note: if no filename extension is specified, Hypersignal defaults to an .fir extension, which specifies FIR Filter Construction output. If a .FIR file isn't found, then Hypersignal searches for a .TIM file. If you have both FIR and time domain files with the same filename, then you may want to specify the extension in the filter name field(s) of the Real-Time Convolver menu (e.g. 'delay.tim').

   Filter Name 2 Starting with v1.3 of the Real-Time Convolver, this field allows a second, independent filter to be applied to channel 1 (second channel), which allows two filters on two different channels.

   Sampling Frequency The desired sampling rate, in Hertz. Note that whether the desired rate can be hit exactly depends on the hardware in use. In order to create a sampling clock, often times hardware systems will utilize an integer counter, or a combination of onboard oscillators, which limit the number of possibilities. In all cases, the Hypersignal Analog Conversion function will display the actual sampling rate used when the Real-Time Convolver is running.

   FFT Order This is an important field. In order to ensure the Real-Time Convolver functions correctly, the FFT length (which is the FFT order raised to power of 2) should be made at least twice as large as the maximum filter length. For example, if the Filter Name 1 field specifies a filter you know to be 4000 pts. (or taps), then you should enter an FFT order of 11 (FFT length of 8192). For optimum Real-Time Convolver efficiency, which is required at high rates and/or stereo operation, the FFT length should be about from 6x to 10x the maximum filter length. There can be exceptions to these guidelines when using very low sampling rates. For more information on filter and FFT lengths, see the section "MAXIMUM IMPULSE RESPONSE LENGTH and SAMPLING RATE", below. If you have entered a filter name specifying a filter whose length exceeds the FFT length corresponding to the entered FFT order, the Real-Time Convolver will cut the filter length in half (use only the first half of your filter). Also, the FFT order should be in a range from 8 to 17 (length 256 to 131,072); results are undefined for order values greater than 17 (note: this is because 32-bit floating-point DSP chip precision limits make higher FFT orders difficult to achieve).

   Zero dB Frequency In version 2.0, this field will allow you to specify the frequency value at which you wish the corresponding magnitude value of impulse response to be considered as zero dB. For standard Hypersignal-generated FIR filters, this field will be ignored, since the passband/stopband characteristics of such filters are known ahead of time, and the coefficients scaled accordingly. Currently, for arbitrary time domain waveform file impulse responses, a root-mean-square scaling factor is calculated based on Parseval's theorem, in order to not add any power to the output data stream as a result of convolution. For more information on scaling, this type of scaling, see the Hypersignal-Macro Users Manual, section 4.9.2, Applying Filters with Convolution.

4. After all input fields are entered, press ctrl-Enter, the F3 key, or the Enter key while the cursor is on the last field. This will cause the Real-Time Convolver macro to run. Note that if you are running the macro for the first time, it will be compiled first.

5. The Real-Time Convolver macro first temporarily configures your System Configuration menu for the "rtcXXX" dsp file, where XXX indicates the type of DSP being used. It then sets up the Analog Conversion menu with the appropriate channel list, filter filename(s), and sampling rate.

6. At this point, you should see the Analog Conversion menu appear, with the "BEGIN ACQUISITION (Y/N)?" prompt displayed at the bottom. Upon answering this prompt with the Y key, real-time convolution should begin. Note that you can stop the convolution using the Esc key, change menu parameters such as filter name and sampling rate, and re-start at any time. The INP,OUT scaling factor field is active also, which allows you to apply digital gain/attenuation independently to the input and output (this field is initialized to '1,1'). Pressing the Esc key again will exit to the main menu, which will cause the macro to resume, restore the System Configuration menu, and terminate.

7. Once real-time convolution is running, please keep in mind the following notes:

   - No data is being acquired to memory or disk. The only way to verify operation is to put a signal into the analog input(s) on the DSP/Analog board, and check (listen) to the analog output(s).

   - Because of the real-time nature of the convolver, long input/output buffers are used to build up sufficient data for processing. This causes long delays from signal input to signal output***, in addition to delay caused by the filter(s)*** (see section "LATENCY CONSIDERATIONS", below). If you are using a microphone setup with live voice input, this can be distracting. You may want to consider either eliminating conflicting visual cues, such as who is speaking and when, or using only recorded data and/or headphones.

MAXIMUM IMPULSE RESPONSE LENGTH and SAMPLING RATE

The upper limit on the FFT Order field above (which implies an upper limit on impulse response length), and maximum sampling rate possible depend mainly on two factors: DSP board memory, and DSP board speed.

1) DSP board memory. The Real-Time Convolver is a DSP memory-hog. There is no way to avoid this unsavory characteristic. Thus, in many cases, not having enough memory can be a real problem. In general, with most DSP boards, the following equations can be used to determine the maximum FFT and filter sizes possible (N is FFT length, M is filter length, F is number of filters, and C is number of channels):

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N(F + 2C + 1) - MC + code_size
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which expresses the amount of memory required in 32-bit words. "Code_size" is an approximate size of program code that varies between processor types; this value accounts for the program size of the convolver itself, and is typically less than 2048. Note that for a floating-point DSP board with as much as 64k x 32 of memory, the above equation indicates a maximum FFT length of only 16384, regardless of other factors. Another way to look at this is to rearrange the equation. First, rewritng the equation as

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N(F + 2C + 1) - MC + code_size  <= max_mem
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and then, using the approximate relationship of N=2*M, an expression can be found for the largest approximate N:

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N <= (max_mem - code_size) / (F + 3C/2 + 1)
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Note that the relationship N = 2*M is a minimum requirement; in otherwords, the FFT size chosen must be at least twice the filter (impulse response) size. For optimum performance (high sampling rates), even longer FFT sizes may well be needed. For low sampling rates, N = 1.5*M or similar might be good enough. There are some examples below given for 44.1 kHz stereo operation, with additional estimates for mono operation and lower sampling rates extrapolated. Note that using two channels instead of one (stereo operation), and two independent filters instead of one, cuts the maximum N almost in half. Also, remember that any maximum value of N should be rounded down to the nearest power of 2, since only power of 2 FFTs are used in the Signalogic Real-Time Convolver.

Some typical maximum memory amounts available, along with conservative estimates of maximum filter length for stereo operation with one filter at the rates given:

- Ariel DSP-32C, 64k x 32, FFTmax = 8192, filtmax = 1200, rate = 44.1 kHz
- CAC QUANTUMdsp DSP3210, 512k x 32 (DRAM), FFTmax = 65536, filtmax = 10000, rate = 12 kHz
- Signalogic Sig32C-8, 128k x 32, FFTmax = 16384, filtmax = 2480, rate = 44.1 kHz
- National Instruments AT-DSP2200, 384k x 32, FFTmax = 65536, filtmax = 10000, rate = 44.1 kHz
- Signalogic Sig32C-4, 640k x 32, FFTmax = 131072, filtmax =20000, rate = 44.1 kHz

Note that in the above estimates, FFT sizes of about 6.5x the filter size have been used. As noted earlier, to achieve optimum performance--which is definitely required for 44.1 kHz stereo operation--FFT sizes greater than 2x the filter size are typically necessary. For mono operation at 44.1 kHz, the above filter sizes can usually be multiplied 2-3 times. Also, for 80 MHz versions of the DSP32C, the above filter sizes can likewise be increased.

Please see the Signalogic hardware brochure "Hypersignal-Macro: Putting Hardware to Work" for memory size specifications for various boards.

2) Board Speed. The maximium sampling rate possible with any given boardis largely independent of the maximum filter size, which is explained above.The factors that determine maximum sampling rate are mainly DSP chip speed, and access time (speed) of onboard SRAM. Boards that contain DRAM, whileoffering long filter lengths, will be much slower because the individual DRAM devices are slower than their SRAM counterparts.

3) Putting the two together. It has been our experience in working with acoustic customers that often times a given filter length is absolutely required. For example, if you need 400 msec of impulse response at 44.1 kHz, then your filter length must be at least 17640 samples long. If this exceeds the maximum memory limitations of the DSP board, then you must decide: which is more important, the 400 msec time period or the 44.1 kHz sampling rate? Assuming that the 400 msec is needed, then you can reduce the sampling rate, and perform a frequency decimation on the impulse response (or use some other method) to reduce its length as well. The results won't be as precise as with a 44.1 kHz sampling rate, but they may well be a very close approximation.

In cases where memory is not a limiting factor, maximum sampling rates are determined by the speed considerations mentioned above, and by the number of channels required. An AT&T DSP3210 with a 60 nsec instruction cycle time, for example, will be faster than an AT&T DSP32C with an 80 nsec instruction cycle time. Note that at lower sampling rates, when you are not "pushing the envelope" of maximum sampling rate performance, you can "violate" the 2x minimum relationship of FFT length to impulse response length mentioned above--to a certain extent. For example, at a 12.5 kHz rate, it may be possible to run a 7000 pt. impulse response length with an 8192 pt. FFT length. However, at ***no time*** should you attempt to download a filter that is longer than the FFT length--results are undefined. Again, you will have to experiment to determine the maximum rate possible given your desired setup conditions.

SOURCE OF FILTER COEFFICIENTS

Any waveform file from any source that can be imported into Hypersignal can provide filter coefficients. The Hypersignal-Macro family of software packages contain a mature, flexible "File Acquisition" function that is designed to import waveform data from a variety of formats and data types. Also, Hypersignal-Acoustic includes "beta" versions of macros that convert ETC files from Techron Sound Lab software format and from JBL CADP2 software format directly into Hypersignal time domain format (.tim file).

SYSTEM LATENCY

Because of the digital (DSP-based) nature of the Real-Time Convolver, an inherent latency is present in the system, even if you download a "zero-delay" set of coefficients that consists of an initial "one" followed by zeros. This latency can be calculated as follows:

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2*(FFT length - filter length)/sampling rate
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where sampling rate is in kHz, and latency is in msec. Clearly, you can reduce the system latency for a given filter size by reducing the FFT length. However, the circumstances under which this is possible and real-time operation can still be maintained are limited, as described above.